Dakota
User's Guide

199 Heater Road • Lebanon, NH 03766 • 603-448-6283

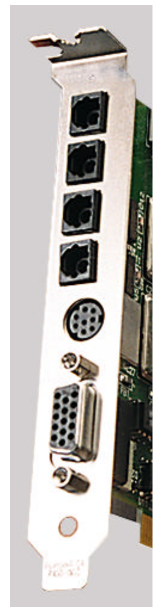
FRONTIER DESIGN GROUP

Contents

continued➔

≡ Introduction ≡

Dakota™ is a PCI soundcard that integrates many digital audio, MIDI, and synchronization features:



- Two optical ports for digital audio input (16 channels, ADAT optical format)
- Two optical ports for digital audio output (16 channels, ADAT optical format)
- One 8-pin port with a breakout cable for 2 MIDI inputs and 2 MIDI outputs
- One 15-pin port with a breakout cable for ADAT 9-pin sync input and SPDIF digital audio (2-channel input and 2-channel output on RCA connectors)
- Frontier Design Group's innovative SoDA™ technology (SMPTE on Digital Audio) allows any audio input or output to transmit SMPTE timecode without using dedicated SMPTE I/O ports

Any optical port can be switched from multichannel ADAT optical format to stereo SPDIF format for connectivity with DAT machines, MiniDisc recorders, or any other TOSLINK-compatible equipment.

Dakota also includes a cable that lets you transfer digital audio directly from a CD-ROM drive, provided that the drive has a compatible 2-pin digital audio output.

Dakota supports 44.1 and 48 kHz sample rates, and dynamically resamples 8, 11.025, 16, 22.05 and 32 kHz digital audio for playback through high-quality external converters. Dakota can also varispeed and lock to digital audio inputs from 39–51 kHz.

Dakota provides software controls for digital audio input status, sample rate, sync source, and timecode format, source and destination. It is also bundled with Cool Edit Pro SE, a multichannel recording and editing application from Syntrillium Software.

Dakota supports sample-accurate audio transfers when used with applications that also support sample accuracy, such as Syntrillium's Cool Edit Pro and Steinberg's Cubase VST.

Dakota's ASIO driver provides ultra-low latency when adjusting parameters and monitoring inputs with applications such as Cubase VST.

Installing Frontier Design Group's Montana™ card along with Dakota expands your system to 32 audio inputs and 32 audio outputs on ADAT optical ports, and adds ADAT 9-pin sync output and an RCA jack for video sync or word clock input.

The ADAT optical ports on Dakota and Montana can be connected to any device that also has ADAT optical ports, including digital mixers, ADAT-compatible tape machines, digital format translators (such as the Apogee FC-8 for TDIF/ADAT translations), and external A/D & D/A converters such as Frontier Design Group's Tango24™ and Zulu™.

Dakota's MIDI capabilities can also be expanded to 8 MIDI inputs and 8 MIDI outputs by connecting its 8-pin port to Frontier Design Group's Sierra™ box, which also has dedicated jacks for SMPTE timecode I/O.

In addition to this User's Guide, your Dakota package should contain:

- Dakota PCI circuit board
- Breakout cable for 2 MIDI inputs and 2 MIDI outputs (four standard MIDI 5-pin jacks); connects to Dakota's 8-pin circular connector
- Breakout cable for ADAT 9-pin sync and SPDIF input and output (a 9-pin jack and two RCA jacks); connects to Dakota's 15-pin D-sub connector
- Internal cable for CD-ROM digital audio
- Envelope containing Dakota software on a 3.5" floppy disk (please read the envelope before opening it)
- CD-ROM with Cool Edit Pro SE recording/editing software and demo versions of industry-leading audio production software

About the Company

Frontier Design Group develops, manufactures and sells digital audio hardware and software. Our goal is to provide high-quality, high-value tools to help our customers be more creative and productive.

We're always interested in receiving your feedback on our products, as well as your suggestions for improvements and new products.
You can send e-mail to feedback@FrontierDesign.com or send a fax to 603-448-6398.

Technical Support

If you have any problems or questions that aren't addressed in this Guide, there are three ways to get more help:

- Refer to our web site (http://www.FrontierDesign.com) for information on current revisions, answers to frequently asked questions, troubleshooting procedures, and additional documentation. The web site is available every day, 24 hours a day.
- Send specific questions via e-mail to support@FrontierDesign.com. We do our best to respond promptly (usually within one business day).
- For time-critical questions, you can call Frontier Design Group at 1-800-928-3236 (outside the USA, call 603-448-6283). Phone support is normally provided weekdays from 9:30 am to 5:30 pm EST.

≡ Installation ≡

WARNING! The components in your computer and on the Dakota board are sensitive to electrostatic discharge. Follow these precautions to avoid damage caused by static electricity —

- Leave the Dakota board in its anti-static wrapping until you're properly grounded.
- To become grounded, make sure the computer is off but leave its power cord plugged in. Remove the cover, and touch the computer's metal chassis.
- Only handle the Dakota board by its edges and metal bracket.

Installing Dakota

Dakota requires one PCI slot in your computer. If you're planning to install Montana in addition to Dakota, be sure to install Dakota in a slot that is adjacent to an empty PCI or ISA slot.

1. Remove the protective plugs from Dakota's four optical ports.
2. Make sure the computer is off (but leave its power cord plugged in), and then remove its cover.
3. Using a screwdriver, remove the blank metal bracket from an empty PCI slot in the computer, and verify that the motherboard has no protruding components that would obstruct the 15-pin connector near the bottom of the Dakota bracket.
4. Insert the Dakota board into the empty PCI slot, and secure the Dakota bracket with a mounting screw.
5. If your computer has a CD-ROM drive with a 2-pin digital audio output, you can connect it internally to the Dakota board using the CD-ROM cable included with Dakota. Push one end of the cable into the CD-ROM drive's output connector until it clicks into place.

6. Route the CD-ROM cable inside the computer to Dakota, and push the other end of the cable into Dakota's internal CD-ROM connector (labelled CD-ROM) until it clicks into place.

> Note: Not all CD-ROM and DVD drives have a 2-pin digital audio output, and some have the connector but it isn't active (such as many Memorex drives).
>
> For a list of compatible drives, you can refer to the Dakota Frequently-Asked-Questions (FAQ) page on our web site (http://www.FrontierDesign.com).

7. Replace the cover, and restart your computer. As it starts up, Windows detects the presence of the new hardware, and automatically opens an installation wizard. In Windows 95, it's the "Update Device Driver" wizard. In Windows 98, it's the "Add New Hardware" wizard.
8. Follow the instructions in the wizard, letting Windows search for the best driver for your device. You'll need to insert the Dakota Software disk into the floppy drive and click the "Next" button as the wizard leads you through the driver installation process.
9. When the wizard indicates that the Dakota Audio Device is installed, click the "Finish" button.
10. After the installation is finished, remove the floppy disk and keep it in a safe place in case you ever need to re-install Dakota.

> Note: Dakota ships with its MIDI ports disabled (to avoid a Windows 95 bug that crashes the system if more than 11 MIDI ports are active). For details on enabling Dakota's MIDI ports, please refer to the 'System Tab' section of the Software Reference chapter later in this Guide.

Your Dakota shipped with the most recent software available at the time. The latest version of Dakota software is always available on our web site (http://www.FrontierDesign.com), and you can download it for free.

Installing Cool Edit Pro SE

Dakota comes with a copy of Cool Edit Pro SE, a multichannel recording and editing application created by Syntrillium Software.
Cool Edit Pro SE is easy to learn and fun to use.

To install Cool Edit Pro SE:

1. Insert Dakota's CD-ROM into your computer's CD-ROM drive. Your browser will automatically open the Dakota Software Sampler home page from the CD-ROM.

   (If your browser doesn't automatically open the CD-ROM, you can open it manually by navigating to the "\Frontier\Default.htm" file on the CD-ROM and double-clicking on it.)

2. When the home page appears, follow its instructions to install Cool Edit Pro SE and the demo versions of other software.

3. After the installation is finished, remove the CD-ROM and keep it in a safe place in case you ever need to re-install Cool Edit Pro SE.

   This application is a limited version of Cool Edit Pro. The SE version supports only 10 (mono or stereo) tracks of audio and has no DSP effects. The CD-ROM includes details about how to upgrade your SE version to Cool Edit Pro, complete with 64 audio tracks and more than 30 high-quality DSP effects.

Uninstalling Dakota

To uninstall Dakota, you need to remove the driver and control panel software, and remove the Dakota hardware.

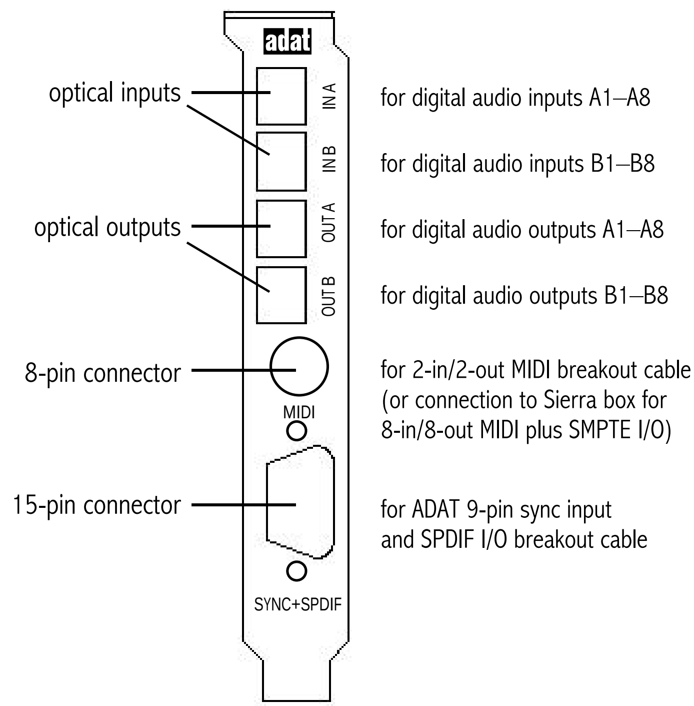
— Removing the software —

1. Open the Windows Control Panel (Start→Settings→Control Panel), and double-click the System icon.
2. When the System window appears, select the “Device Manager” tab.
3. Click the ‘+’ next to “Sound, video, and game controllers” to open that part of the device tree.
4. Select “Dakota Wave and MIDI Driver,” and click ‘Remove.’
5. When prompted to “Confirm Device Removal,” click ‘OK’ to remove the driver software.
6. To remove the control panel software, open the Windows\System folder and delete the “DakPanel.cpl” file.

— Removing the hardware —

1. Make sure the computer is off (but leave its power cord plugged in), and then remove its cover.
2. Disconnect the internal CD-ROM cable from the Dakota board.
3. Remove the mounting screw that holds the Dakota bracket in the computer.
4. Lift the Dakota board straight out of its PCI slot.
5. Insert the plastic protective plugs into the four optical ports, and put the Dakota board into an antistatic bag.
6. Replace the blank metal bracket and the computer’s cover, and restart your computer.

Dakota Connectors



Connecting Dakota to Other Devices

Dakota has several types of connectors on it. Here are some suggestions for using them.

— Optical ports and cables —

Dakota ships with plastic plugs covering its optical input and output connectors. Before installing Dakota in your computer, you can remove the plugs by pulling them out of the connectors.

The optical ports accept standard TOSLINK™ optical cables. You can purchase optical cables in lengths up to 10 meters (33 feet) from Frontier Design Group. Many optical cables have small flexible plastic covers on their tips to prevent scratches. Make sure you remove the plastic covers before connecting the cable to an optical port.

The optical connectors are flat on one edge and rounded on the other, so if the cable doesn't easily snap into the port, rotate the end of the cable 180° and try again.

You can connect either end of an optical cable to any optical port, as long as you're connecting an output to an input. The direction of an optical signal is easy to verify because optical outputs emit red light. The light itself is not dangerous, since TOSLINK and ADAT optical ports use light-emitting diodes (LEDs) rather than laser light.

Dakota's optical ports can be used in either 8-channel or 2-channel mode.

- The 8-channel ADAT mode lets you connect Dakota to multitrack tape machines, digital mixers, multichannel external A/D and D/A converter boxes such as Tango24 and Zulu (also from Frontier Design Group), and other devices that use 8-channel ADAT optical I/O.
- The 2-channel SPDIF mode lets you connect Dakota to DAT machines, CD players, MiniDisc recorders and other devices that use 2-channel SPDIF optical I/O. See the "Software Reference" chapter later in this Guide for more details.

— 8-pin port and cables —

Dakota's 8-pin port is for MIDI I/O. It can be connected either to the MIDI breakout cable that's included with Dakota, or to a Sierra (Frontier Design Group's multiport MIDI and SMPTE I/O box).

Connecting the 8-pin port to Dakota's MIDI breakout cable gives you 2 MIDI inputs and 2 MIDI outputs on standard 5-pin MIDI jacks. Since MIDI inputs and outputs look the same, make sure you check the labels and connect your MIDI cables to the appropriate jacks.

If you want to use Sierra (for 8 MIDI inputs, 8 MIDI outputs, and SMPTE I/O), connect Dakota's 8-pin port to the cable that's included with Sierra instead of connecting Dakota's MIDI breakout cable, and make sure you enable Sierra in the Dakota control panel's System tab. See the "Software Reference" chapter later in this Guide for more details.

— 15-pin port and cable —

Dakota includes a breakout cable that connects to its 15-pin port. When you attach the cable to Dakota, make sure you tighten the jack screws on each side of the connector.

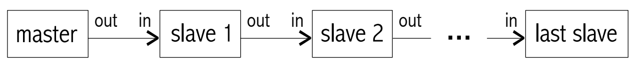
> WARNING! Do not plug your video monitor into Dakota's 15-pin port. Although a monitor cable can physically fit into the connector, it is not intended for that purpose.

The breakout cable has a DB-9 connector for ADAT 9-pin sync input, and two RCA (phono) jacks for SPDIF coaxial I/O. Since the input and output jacks are the same size and shape, make sure you check the labels and connect your RCA cables to the appropriate jacks.

• ADAT 9-pin sync

Dakota's ADAT sync input can be connected to any device with an ADAT sync output. To insure proper data transmission, use only cables that are made for ADAT sync connections.

ADAT sync connections transmit audio clock (sample rate), timecode, and machine control commands down a sync "chain," and responses to the commands run up the chain. The chain runs from the sync output of the first device (the master) to the sync input of the next device (slave 1), and from the sync output of slave 1 to the sync input of slave 2, and so on.



> Note: If you've installed Dakota but not Montana, make sure the computer is the last slave device in your ADAT sync chain. If you've installed both Dakota and Montana, the computer can be in any slave position in the chain.

For details about sample-accurate transfers, see 'Dakota Sync Features' in the "Synchronization" chapter later in this Guide.

• SPDIF cables

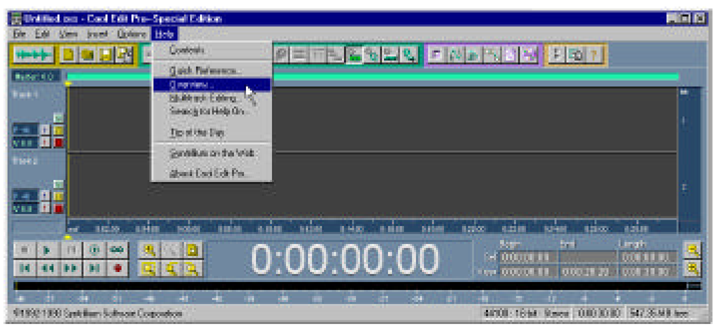
Standard audio cables have twisted-pair wire rather than coaxial cable, and are not recommended for use with SPDIF (Sony/Philips Digital Interface Format) signals. Although they have the correct connector on each end, and may work over short distances, they are not reliable for SPDIF data streams. The reason is related to the differences between audio and SPDIF signals — audio signals contain frequencies in the tens of kiloHertz, but SPDIF signals contain frequencies in the tens of megaHertz, much like a video signal.

> For reliable SPDIF connections, especially over distances longer than a meter, you should use 75-ohm coax video cables.

≡ Tutorial ≡

This tutorial uses Cool Edit Pro SE to demonstrate some of Dakota's features. During installation, you probably created a Cool Edit Pro program group. If so, you can open the application from the Start menu (Start→Programs→Cool Edit Pro→Cool Edit Pro Special Edition).

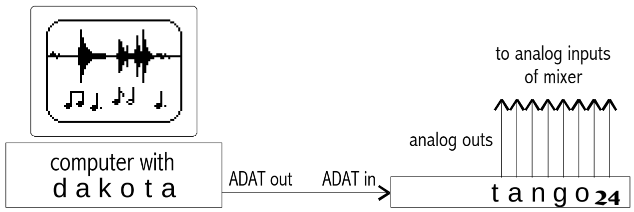
If you haven't used Cool Edit Pro before, please become familiar with it first. To learn from its online documentation, select 'Overview…' from the Help menu, click the 'Contents' button and open a topic such as 'Getting Started' or 'Navigating Cool Edit Pro.'



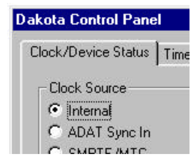
Preparing for Playback

This tutorial assumes you're using a Frontier Design Group Tango24 as a D/A converter for Dakota, but any external converter can be used instead. If you don't have a Tango24, see "Using External Devices for Playback" later in this section for other setup suggestions.

1. Connect an optical cable from Dakota's "OUT A" port to the Tango24 optical input.
2. Set the Tango24 clock switch (on the front panel) to 'Optical' so the Tango24 will be slaved to Dakota's optical output.
3. Make sure the Tango24 analog outputs are properly connected to your monitoring system.



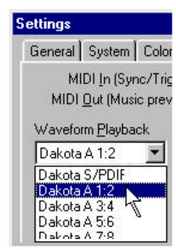
4. To open Dakota's control panel from the Start menu, navigate to Settings→Control Panel, and double-click on the "Dakota" icon.



5. In the Dakota control panel, make sure the Clock Source is set to 'Internal' (so Dakota is the sample clock master) and set the Sample Rate to '44.1kHz.'

Playing a Sound

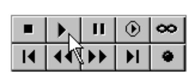
1. Open Cool Edit Pro SE (CEP) and make sure you're in the 'Edit Waveform' view. (F12 toggles between the 'Edit Waveform' and 'Multitrack' views.)



2. Go to Options→Settings… and open the Devices tab.

3. Set the 'Waveform Playback' field to the desired stereo output (for example, 'Dakota A 1:2' to use Dakota/Tango24 outputs 1 & 2) and click 'OK.'

4. Open any sound file (.wav file) on your system. If you don't have a favorite sound file, you can go to File→Open… and navigate to C:\Windows\Media, where you should find 'The Microsoft Sound.wav'. When you double-click on the sound file, its waveform should appear in the CEP window.



5. Click the Play button (near the bottom left of the CEP window) to hear the sound. When the sound is playing, the 'Clock/Device Status' tab in Dakota's control panel should show bright green status icons for digital outputs A 1:2.

Using External Devices for Playback

You can use a variety of playback devices to listen to Dakota's outputs. Here are some general guidelines for several types of external devices. In all cases, the best setup for tutorial playback has Dakota as the sample clock master at 44.1kHz, and the external device slaved to its digital input.

• External Converters — Frontier Design Group's Zulu is always slaved to its optical input, so you just need to connect Zulu to the correct Dakota output and set Dakota's clock source to 'Internal' and sample rate to 44.1kHz. If you're using a 2-channel external converter, make sure you select SPDIF (rather than an ADAT pair) as your output (playback) device. Refer to the "Software Reference" chapter later in this Guide for details on using the optical ports for 2-channel output.

• ADAT-Compatible MDMs — To use an Alesis, Panasonic or Fostex multitrack deck as a D/A converter, set it to 'All Input' monitoring, 'Digital' input, and 'Dig 44.1kHz' clock source. On an original (black faceplate) ADAT machine, enabling 'Digital In' automatically sets the clock source to its optical/digital input. On an ADAT-XT, use the Clock Select button to choose 'Dig 44.1kHz.'

• DAT Machines — To use a DAT machine as a D/A converter, set its input source to 'Digital' and enable its 'Record-Pause' mode to monitor the digital inputs. Make sure you select SPDIF (rather than an ADAT pair) as your output (playback) device.

The input monitoring function works better on some DATs than on others. To enable input monitoring, a DAT may:

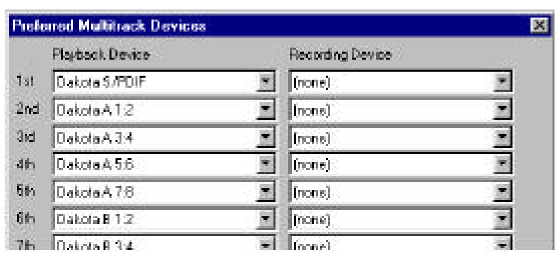
– not need to have a tape loaded (ideal)
– need to have a tape loaded, but not need to spin up its head (OK)
– need to have a tape loaded, and need to spin up its head (unfortunate)

The last case is particularly undesirable because the spinning head will unnecessarily wear both the tape and the head. If you're considering purchasing a DAT machine, ask whether it must have a tape loaded and whether it needs to spin up its head to enable input monitoring.

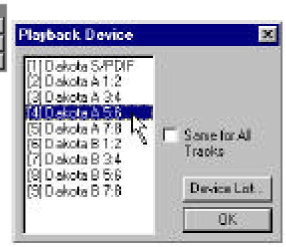
Note: Unlike Tango24 and Zulu, you cannot simultaneously use both the A/D and D/A converters in a DAT machine or an ADAT-compatible MDM. To record and play back simultaneously, you'll need to use separate external recording and playback devices.

Editing the Sound

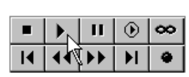
1. In the CEP window, go to Edit→Select Entire Wave, and then Edit→Copy.
2. Click at the end of the waveform, and then go to Edit→Paste.
3. Go to View→Multitrack View, and then File→New Session… and make sure that 44100, Stereo, and 16-bit are selected. Click ‘OK.’
4. Click near the beginning of Track 1 to indicate where you’ll be inserting the edited waveform.
5. Go to the Insert menu and select the name of your file. Your edited waveform should appear on Track 1.
6. Go to Options→Device Preferences… and map the CEP Playback devices to Dakota, as shown here. Click ‘OK.’



7. In the control area at the left of Track 1, click the grey button with the blue number on it to open the ‘Playback Device’ window.



8. Select the desired output device (‘Dakota A 5:6’, for example) and click ‘OK.’ (This is how you route tracks to Dakota output channels in CEP.)
9. Click the Play button to playback Track 1. You should hear the sound play twice.



Following SMPTE or ADAT Sync

If you have a source of SMPTE timecode or ADAT Sync, you can use it to control CEP and play the sound.

1. In Dakota's control panel, set the Clock Source field to 'SMPTE/MTC' or 'ADAT Sync' to indicate which type of control signal you're using.
2. If you selected 'SMPTE/MTC,' open the Timecode tab and set the Timecode Source field to your SMPTE input (see "SMPTE I/O" below for details). Also make sure the frame rate matches the incoming timecode.
3. In CEP's Multitrack View, go to Options→SMPTE Start Offset… and select the frame rate that matches the setting in Dakota's control panel. (Set an offset only if you aren't starting at time 0).
4. Go to Options→Settings…→Devices, set the MIDI In (Sync/Trigger) field to 'Dakota 1' and click 'OK.'
5. On the Options menu, select 'SMPTE Slave enable' — a check mark should appear next to that item in the menu.
6. Start the timecode source to play the sound under external control.
7. When you finish, go to Options→SMPTE Slave enable again to turn off timecode control — the check mark should disappear from the menu.

SMPTE I/O

Dakota features Frontier Design Group's SoDA™ (SMPTE on Digital Audio) technology — any audio input can be used for receiving SMPTE timecode, and any audio output can be used for sending SMPTE timecode. Dakota also provides integrated SMPTE chase-lock and generation.

If you've connected Frontier Design Group's Sierra MIDI/SMPTE box to Dakota, you can use its dedicated SMPTE I/O jacks to send and receive SMPTE timecode instead of using SoDA channels.

You can select the SMPTE timecode input in the 'Timecode' tab of Dakota's control panel.

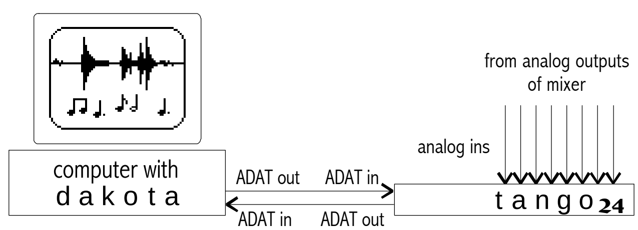
Setting Up to Record

You can record additional tracks in your CEP session by transferring digital audio from an ADAT, DAT or CD player. Or you can record live audio through an external A/D converter, or by using the A/D converters in an ADAT or DAT machine.

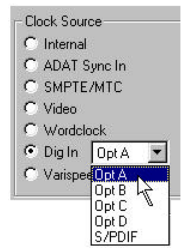
> Note: If you only have one digital tape machine, you'll be able to record during this part of the tutorial, but without hearing the sounds recorded previously (because you can't simultaneously use the A/D and D/A converters in an ADAT or DAT machine). To record and play back simultaneously, you'll need two tape machines or an external converter such as Tango24 or Zulu.

This tutorial assumes you're using a Frontier Design Group Tango24 as an A/D converter for Dakota, but any external converter can be used instead. If you don't have Tango24, see "Using External Devices for Recording" later in this section for other setup suggestions.

1. Connect the Tango24 optical output to Dakota's "IN A" port.
2. Set the Tango24 clock switches to "Internal 44.1kHz" so Tango24 is the sample clock master.

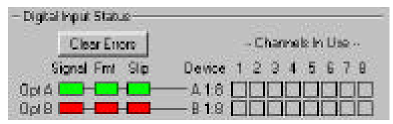


continued➔

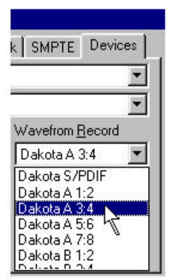


3. In the 'Clock/Device Status' tab of Dakota's control panel, set the Clock Source to 'Dig In Opt A' so the computer is slaved to the Tango24 optical output.

4. Verify that all three of Dakota's 'Opt A' input indicators are green (meaning that the signal is present, it's the correct format, and Dakota's clock is locked to the input). If any indicator is red, refer to the "Software Reference" chapter later in this Guide.



5. In CEP, use F12 or the View menu to select 'Edit Waveform' view. (We'll use 'Edit Waveform' view to check the input level, and then we'll use 'Multitrack' view to actually record.)



6. Go to Options→Settings…→Devices, set the 'Waveform Record' field to the appropriate input (Dakota A 3:4 to use Tango24/Dakota inputs 3 & 4, for example), and click 'OK.'

7. Open the View menu and make sure 'View Level Meters' is checked.

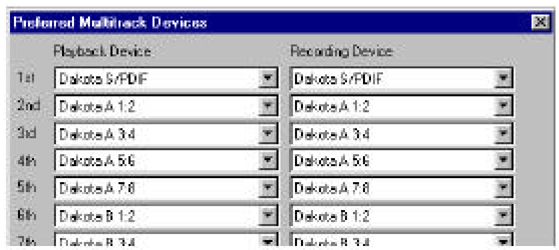
8. Right-click on the horizontal meters (at the bottom of the CEP window) and select 'Start/Stop Meter' from the pop-up menu.

9. Connect the instrument or microphone to the appropriate input, and adjust the input level so it's hot without clipping. When the level is set, select 'Start/Stop Meter' again or click the Stop button to turn off metering.

Recording a Track

1. Use F12 or the View menu to select 'Multitrack' view.

2. Go to Options→Device Preferences… and map the Dakota inputs to the CEP Recording devices, as shown here. Click 'OK.'

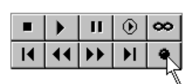


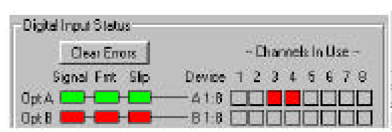
3. Click the Track 2 grey button with the red number on it to open the 'Recording Device' window.

4. Select the appropriate input device (Dakota A 3:4, for example), click 'Stereo' to indicate that we'll be recording both input channels, and click 'OK.' (This is how you route inputs to tracks in CEP.)

5. To record-enable Track 2, click its red 'r' button.

6. Click the Record button (near the bottom left of the CEP window) to begin recording, and play the tape or perform live. While recording, the input status icons for Opt A 3:4 in Dakota's control panel should be bright red, and you should also hear the audio on Track 1.





7. Click the Stop button when you finish recording. In a moment, the new waveform should appear on Track 2.

8. To disable recording on Track 2, click its red 'r' button again.

Using External Devices for Recording

You can use a variety of devices to record through Dakota's inputs. Here are some general guidelines for several types of external devices. In each case, connect the external device's digital output to one of Dakota's audio inputs. During setup, look at the input status indicators (in the Audio tab of Dakota's control panel) to verify that you're receiving proper digital input.

• External Converters — External A/D converters should be set for 'Internal 44.1kHz' operation for this tutorial. If you're using Frontier Design Group's Zulu, leave the Dakota control panel set to 'Internal 44.1kHz' and Zulu will automatically lock to Dakota's sample clock. If you're using a 2-channel external converter, make sure you select SPDIF ('S 1:2' rather than an ADAT pair) as the input device.

• CD-ROMs — Some CD-ROM drives have a 2-pin digital audio output. If you connected such a drive in your Dakota installation, you can select it as the 'S 1:2' input in Dakota's control panel. Some CD-ROM drives only have active output if an audio CD is loaded and playing in the drive, so make sure a CD is actually counting playback time before checking the input status indicators in Dakota's control panel. To use Cool Edit Pro's CD transport controls, go to View→Show CD Player.

• ADAT-Compatible MDMs — To use an Alesis, Panasonic or Fostex multitrack deck as an A/D converter, set it to 'All Input' monitoring, 'Analog' input, and 'Int 44.1kHz' clock source. An original (black faceplate) ADAT machine automatically switches to internal sync when its 'Digital In' button is off. If you're using a BRC or an original ADAT machine, you set the clock to 44.1kHz by pitching down to -147 cents.

• DAT Machines — To use a DAT machine as an A/D converter, set its input source to 'Analog' and enable its 'Record-Pause' mode to monitor the analog inputs. Make sure you select SPDIF (rather than an ADAT pair) as your input (recording) device.

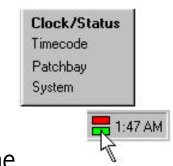
The input monitoring function works better on some DATs than on others. For details, see 'DAT Machines' in the section "Using External Devices for Playback" earlier in this tutorial.

Note: Unlike Tango24 and Zulu, you cannot simultaneously use both the A/D and D/A converters in a DAT machine or an ADAT-compatible MDM. To record and play back simultaneously, you'll need to use separate external recording and playback devices.

≡ Software Reference ≡

To open Dakota's control panel from the Start menu, navigate to Settings→Control Panel, and double-click on the "Dakota" icon.

Dakota's control panel includes four tabs — Clock/Device Status, Timecode, Patchbay, and System. Each tab contains several sections, each with one or more fields.

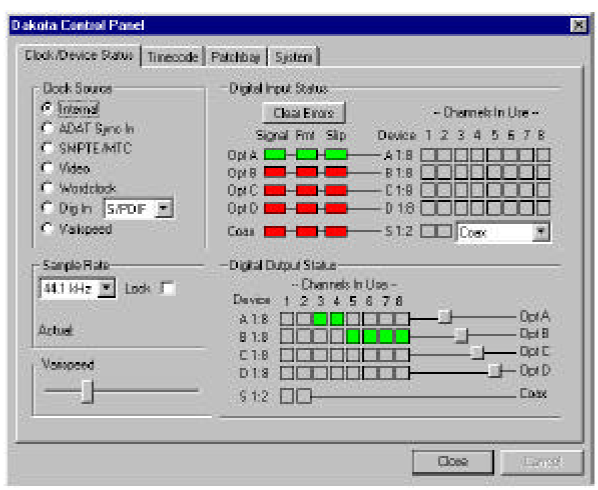


Whenever Dakota is installed in your system, an icon with two bars appears in the Windows Taskbar. The top bar lights red whenever any Dakota input is active, and blinks whenever an active input has an error (see the Clock/Device Status tab for details about any input error). The bottom bar lights green whenever any Dakota output is active.

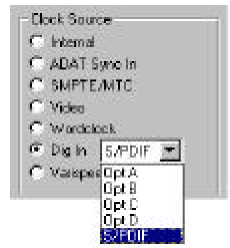
Double-clicking on the icon opens the Clock/Device Status tab. You can also right-click on the icon to display a menu that lets you go directly to the Clock/Status, Timecode, Patchbay, or System tab.

Clock/Device Status Tab

The Clock/Device Status tab contains five sections — Clock Source, Sample Rate, Varispeed, Digital Input Status, Digital Output Status.



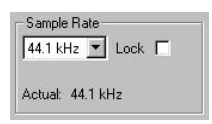
— Clock Source —



The audio clock can come from many sources if you're using Dakota, and even more sources if you're also using Montana.

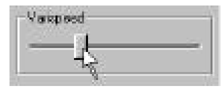
Internal	44.1 or 48kHz, depending on the Sample Rate selection
ADAT Sync In	Dakota can follow word clock from its ADAT Sync In port
SMPTE/MTC	In this mode, Dakota sets its sample clock to match the incoming timecode as selected in the Timecode Source section
Video	With a video source connected to Montana, Dakota can derive a sample clock that is locked to the incoming video signal
Wordclock	With a word clock source connected to Montana, Dakota can set its sample clock to match the incoming word clock signal
Dig In	Any digital input can be used to set the sample rate. 'Opt A' and 'Opt B' are Dakota's optical inputs. 'Opt C' and 'Opt D' are Montana's optical inputs. 'SPDIF' is Dakota's RCA (coax) input.
Varispeed	Lets you use the Varispeed slider to adjust the sample rate

— Sample Rate —



In this section, a drop-down box lets you select the expected sample rate (44.1 or 48kHz). If the 'Lock' box is checked, then applications can only use that sample rate. For example, if the sample rate is locked at 48kHz and you try to play a 44.1kHz sound, an error message will appear. The Sample Rate section also displays the actual current sample rate as measured by the computer.

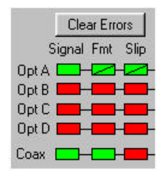
— Varispeed —



If ‘Varispeed’ is selected in the Clock Source field, you can manually set the sample rate to anything between 39 and 51kHz by dragging the Varispeed slider.

— Digital Input Status —

This section displays the audio input status in the form of error indicators, mappings from physical inputs to logical inputs, and channel indicators.



On the left are three rectangular error indicators for each of Dakota’s and Montana’s physical inputs (Opt A, Opt B, Coax, and so on).

A red indicator represents an error, and green means “good to go.” A slash across a green indicator means that at least one error was detected since the errors were last cleared. To remove all slashes, click on the ‘Clear Errors’ button.

From left to right, the indicators are:

Signal
green = input signal active
red = input signal inactive

Format (Fmt)
green = valid digital audio format detected
red = error in digital audio signal format
(for example, feeding an SPDIF signal to an ADAT input)

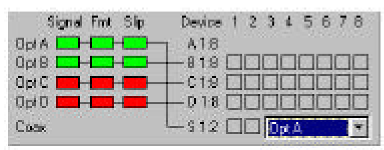
Slip
green = Dakota is locked to incoming audio clock
red = Dakota’s sample rate is different than the incoming audio clock

If Dakota’s sample rate is almost the same as the incoming audio clock, but not perfectly locked, the Slip indicator flashes red periodically. A slash appears across the green indicator until you click ‘Clear Errors.’

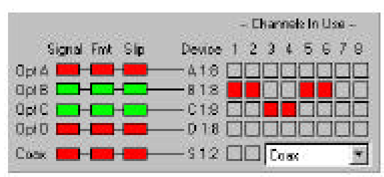
By default, the physical inputs (Opt A, Opt B, Coax, and so on) are mapped directly to the logical devices (A 1:8, B 1:8, S 1:2, and so on) that you'll use within your recording/playback applications.

The stereo device 'S 1:2' (in SPDIF format) can come from any physical input — an optical input, the RCA (coax) input, or the internal CD-ROM 2-pin connector. You can select the stereo input device from the drop-down box next to 'S 1:2.'

The physical-to-logical mapping lines automatically change to graphically reflect your selection.

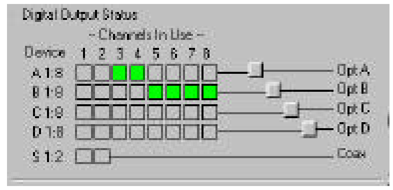


The 'Channels In Use' indicators show activity on each input channel. If an input channel is in use, its indicator is bright red.



— Digital Output Status —

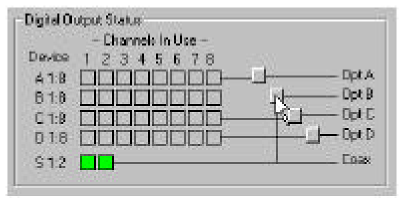
This section shows which output channels are active and the mapping from logical output devices to physical outputs.



When an output channel is playing data, its indicator is bright green.

The stereo SPDIF stream is always present on Dakota's coaxial output (on the breakout cable). You can also route the SPDIF stream to any or all of the Dakota/Montana optical outputs.

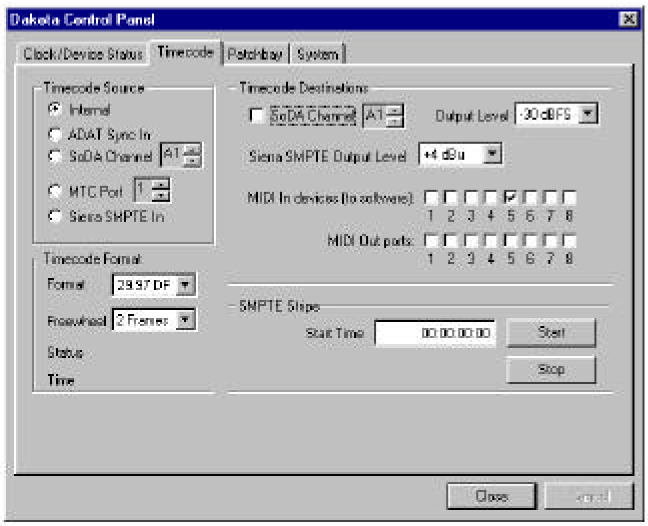
Clicking on the button next to Opt A, B, C, or D toggles the corresponding optical output between 8-channel mode (ADAT optical) and 2-channel mode (SPDIF optical).



The logical-to-physical mapping lines automatically change to graphically reflect your selection.

Timecode Tab

The settings in this tab control where and how timecode is sent.



— Timecode Source —

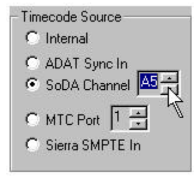
The Timecode Source field lets you select the timecode input. Selecting 'Internal' ignores all external timecode sources, and allows the applications running on your PC to control timecode generation.

Most recording/playback applications can only receive and follow MIDI Time Code (MTC) since it's the only form of timecode in the Windows standard interface. If you're using ADAT Sync or SMPTE input, Dakota automatically converts the timecode to MTC for use by applications.

When 'ADAT Sync In' is selected as the timecode source, Dakota receives timecode information (expressed as a sample count) from the ADAT Sync In port on Dakota's 15-pin breakout cable and passes it to the application in the format specified by the Timecode Format field. In most cases, when the Timecode Source is set to 'ADAT Sync In,' the Clock Source (on the Clock/Device Status tab) should be set to 'ADAT Sync.'

You can also use Frontier Design Group's innovative SoDA™ technology to receive SMPTE timecode on any Dakota or Montana digital audio input —

1. Connect a SMPTE timecode source to an A/D converter (such as Zulu or Tango24), and connect the A/D converter to the desired Dakota or Montana digital audio input.



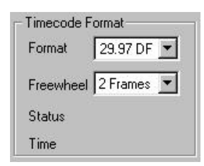
2. Set the Timecode Source field to 'SoDA Channel' and select the appropriate input channel.

To receive MTC from a MIDI input, set the Timecode Source field to 'MTC Port' and select the appropriate port number (1 or 2 if you're using Dakota's 2-in, 2-out MIDI cable, or 1–8 if Dakota is connected to Sierra). See 'Timecode Destinations' later in this section for information about routing the MTC to an application and to MIDI output ports.

To receive SMPTE timecode through Sierra's ¼" SMPTE input, make sure Sierra is properly connected to Dakota, and set the Timecode Source field to 'Sierra SMPTE In.'

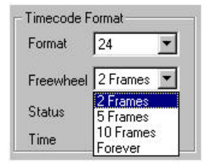
— Timecode Format —

The Format field specifies the form of timecode (SMPTE or MTC) expected to be received and generated. The timecode rate, expressed in frames per second (fps), can be set to:



- 24 – for film applications
- 25 – for PAL or SECAM video rates
- 29.97 ND – NTSC input without drop frame (wallclock ≠ timecode)
- 29.97 DF – standard timecode for NTSC video
- 30 ND – for most music projects and NTSC black/white video
- 30 DF – rare (to "pull down" the sample rate if the external timecode source is 29.97fps)

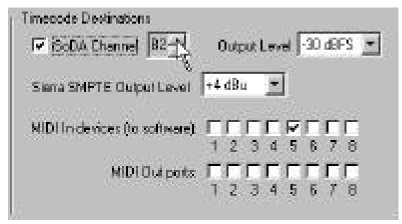
Sometimes a SMPTE timecode signal has dropouts or noise that temporarily garble the timecode. Similarly, a MIDI Time Code packet can be lost. Dakota lets you "freewheel" past these temporary problems in timecode input. The Freewheel setting lets you control how long Dakota continues generating timecode in the presence of poor or idle input.



The Status line indicates when a timecode input is active, and whether Dakota is able to lock to it. The current time appears in the Time field.

— Timecode Destinations —

To use SoDA for sending timecode through a Dakota or Montana digital audio output, check the 'SoDA Channel' box and select the desired output.



You can control the output level on the SoDA channel from -19dBFS to -48dBFS using the Output Level field.

Whenever Dakota is receiving timecode and is properly connected to Sierra, the timecode is automatically sent out from Sierra's SMPTE Out port. Dakota's Output Level field lets you set Sierra's timecode output level for a +4dBu or -10dBV environment.

Incoming timecode can be sent to an application by merging it on a MIDI input port. The 'MIDI In' check boxes let you select any or all MIDI input ports. For example, enabling MIDI In 5 allows applications to sync to Dakota's timecode reader by selecting 'Dakota MIDI 5' as a MIDI input device. Similarly, the 'MIDI Out' check boxes let you send MTC out any of the physical MIDI output ports in your system.

> Note: If Sierra is not connected to Dakota, only MIDI ports 1 and 2 are active. Ports 3–8 can only be used if Sierra is properly connected to Dakota.

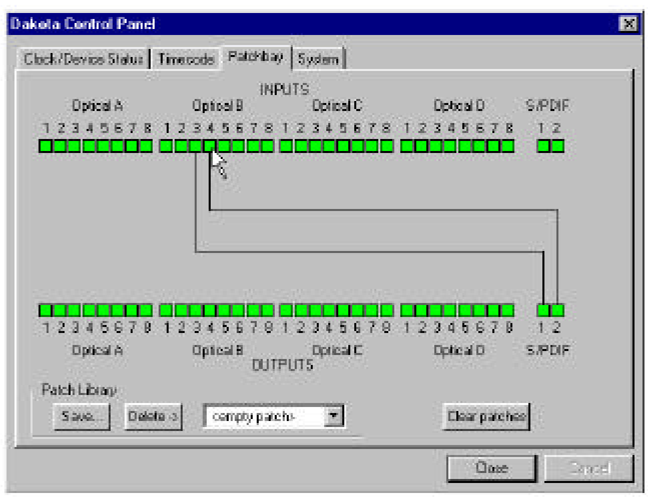
— SMPTE Stripe —

You can use Dakota's timecode generator for striping SMPTE timecode on an external device. Enter the desired time (written as hours:minutes:seconds:frames) in the 'Start Time' field.

When you click the Start button, Dakota's timecode source automatically switches to 'Internal' and SMPTE timecode is sent to the appropriate outputs. Click the Stop button when you finish striping the timecode.

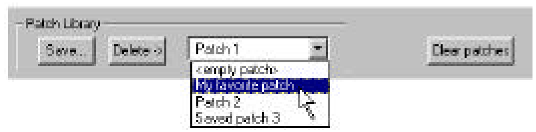
Patchbay Tab

The Patchbay tab lets you route any of Dakota/Montana's audio inputs to any of the audio outputs. You can use the Patchbay tab both for input monitoring and for digital audio format conversions.



Dakota and Montana inputs appear in the top row of icons; outputs appear in the bottom row. To create a patch, click on an input icon and then click on an output icon, or drag the cursor from an input to an output. Lines automatically appear to indicate the routing(s).

For example, you can monitor Dakota's "Optical B 3:4" inputs through the SPDIF coax outputs, as shown here.
To delete a particular routing, double-click on its input or output icon.
To undo all the current routings, click the "Clear patches" button.

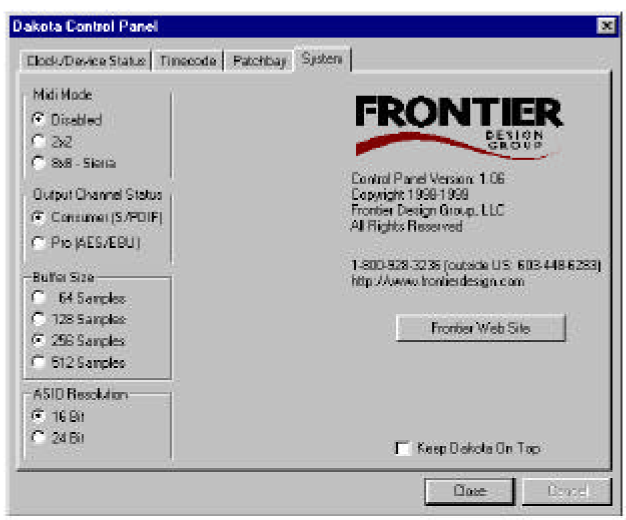


To save the current routings in the Patch Library, click "Save…" and type a name for the patch. You can then restore those routings by selecting the saved patch from the drop-down box.

To remove a saved patch from the Library, select the patch in the drop-down box and then click the "Delete" button.

System Tab

In addition to displaying the Dakota software version and contact information for Frontier Design Group, the System tab lets you enable the MIDI ports and set other options.



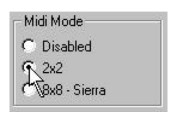
Clicking the 'Frontier Web Site' button launches access to our web site if your PC has an internet link and web browser. The System tab also has a checkbox labelled "Keep Dakota On Top." When the box is checked, the control panel remains in front of other windows on your screen.

— MIDI Mode —

To avoid a Windows 95 limitation (see warning below), Dakota ships with its MIDI ports disabled. Use the MIDI Mode section to enable the MIDI ports for Dakota or Sierra.

> WARNING! Windows 95 is limited to a maximum of 11 MIDI input ports and 11 MIDI output ports. Adding Dakota's (or Sierra's) MIDI ports may put your system beyond the limit, causing it to crash with the "blue screen" message "MSGSRV32 caused a general protection fault…".
>
> There is no easy way to recover from this crash, so before enabling MIDI ports in the Dakota control panel, make sure your Windows 95 system doesn't have too many other MIDI ports enabled (9 if you're using just Dakota; 3 if you're using Sierra). This MIDI limitation is not a problem in Windows 98.

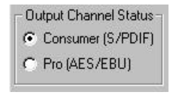
If you're using the Sierra 8-in, 8-out MIDI/SMPTE expansion box with Dakota, select '8x8 Sierra.'



If you're using Dakota's 2-in, 2-out MIDI breakout cable instead, select the '2x2' option.

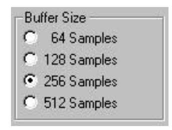
Whenever you change the MIDI Mode setting, you need to reboot the computer for the change to take effect.

— Output Channel Status —



The Output Channel Status section lets you set the channel status bits for Dakota's stereo outputs to either Consumer (SPDIF) or Pro (AES/EBU). The default is SPDIF.
If you have an external SPDIF-to-AES/EBU converter, you can use Dakota's Pro (AES/EBU) setting to generate true AES/EBU output.

— Buffer Size —



The Buffer Size indicates how often data is processed between the Dakota hardware and its driver. Smaller buffer sizes yield faster response times (lower latency), which most noticeably affect audio functions such as monitoring inputs in the Patchbay or using Dakota's ASIO driver in Cubase VST. Larger buffer sizes provide more protection against audio glitches. The default setting is 256 samples. We recommend that you set the buffer size as large as possible while attaining suitable latency for your audio.

> Note: You must close all audio applications before you change Dakota's buffer size. The new setting takes effect when you restart the application(s).

— ASIO Resolution —

You can set the resolution for recording and playback to '16 bit' or '24 bit' when using Dakota's ASIO driver in applications such as Cubase VST. You may also need to set the resolution in the application itself. For example, to record and play 24-bit audio in Cubase VST, you have to enable its '24-bit' button.

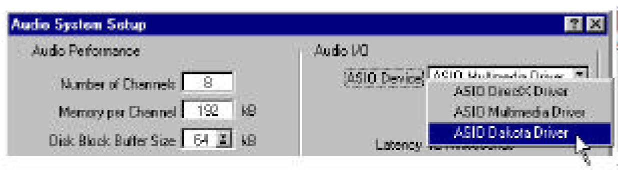
> Note: You must close all audio applications that use Dakota's ASIO driver before you change Dakota's ASIO resolution. The new setting takes effect when you restart the application(s).

ASIO Driver

Dakota's ASIO driver provides ultra-low latency for real-time audio functions such as input monitoring and adjusting EQ, pan, and other parameters.

To use Dakota's ASIO driver in Cubase VST:

1. Close Cubase VST, and open the System tab of Dakota's control panel.
2. Select the desired Buffer Size and ASIO Resolution settings (described on the previous page).
3. Open Cubase VST, and select "System…" from the Audio menu.
4. In the 'Audio I/O' section of the Audio System Setup window, set the 'ASIO Device' field to "ASIO Dakota Driver."



5. Verify that the sample rate, audio source, number of channels, and other options are correct. Using larger disk block buffer sizes in Cubase VST may improve performance (ie, the number of tracks).
6. Click "OK" to close the Setup window.

Note: To actually record or play 24-bit audio, you must set Dakota's ASIO Resolution to '24 Bit' (in step 2) and you also need to enable the '24-bit' button in Cubase VST.

≡ Synchronization ≡

Whenever you transfer digital audio from one device to another, the devices must be properly synchronized. There are two types of synchronization, and each can be transmitted by a variety of signals.

- Sample rate synchronization is required in every digital audio system to guarantee that all devices in the system are recording and/or playing audio at exactly the same rate.
- Timecode synchronization is not always required, but allows two or more devices to reference the same position in time. Timecode facilitates synchronizing a computer with another recorder or sequencer, so that (for example) you can press Play and have all your equipment begin playback exactly three bars into the second chorus.

Sample rate sync

Every digital audio system operates at a particular sample rate, such as 44,100 samples per second (44.1kHz) for CDs. The sample rate can be explicitly transmitted between devices (using BNC word clock, for example) or it can be embedded in a digital audio stream that uses a self-clocking format (ADAT optical, SPDIF, or AES/EBU). The sample rate can also be derived from NTSC or PAL video signals.

When two or more digital audio devices transfer data, one and only one of them should be used to set the sample rate. This device can have a variety of names — system clock source, sync source, sample clock master, audio clock master, or word clock source. The master device uses an internal oscillator to set the sample rate. All other devices must be synchronized (slaved) to the sample clock master, so their digital clocks are locked to the master clock.

No two oscillators run at exactly the same rate, even if they're both 44.1kHz oscillators (for example). Therefore, if two devices in a system are set to internal sync, each using its own internal oscillator, one is inevitably a bit faster than the other. As a result, samples are lost or duplicated during the transfer, corrupting the audio.

When one device (the slave) gets its sample rate from another device (the master), we say that the slave device is locked to the master. In contrast, when two devices are operating at independent sample rates, they are slipping relative to each other. When multiple sample rates are detected, some devices mute, some generate clicks and pops, and some exhibit completely erratic behavior.

> REMEMBER! Any digital audio system must always have exactly one sample clock master, which should determine the sample rate for all devices in that system. If the sample rates don't match, you'll hear glitches, dropouts, or nothing at all.

Dakota and Montana both use Dakota's clock system (and Sierra doesn't need a sample clock), so you can think of them as a single device for sample rate purposes. Tango24 and Zulu, however, are separate devices and must be properly synchronized when connected to either Dakota or Montana.

For example, to use Zulu with Dakota (alone or with Montana), set Dakota's Clock Source to 'Internal 44.1 or 48kHz' so that Zulu is slaved to Dakota.

When using one or more Tango24 converters with Dakota (alone or with Montana), there are many sync options:

- To use a Tango24 as the master, set its clock switches to 'Internal 44.1 or 48kHz.' Set all other Tango24 units to 'Word Clock' (coming from the master Tango), and set Dakota's Clock Source to a digital input that's connected to one of the Tango24 units (Opt A, for example).
- To use Dakota as the master, set its Clock Source to 'Internal 44.1 or 48kHz,' and set all the Tango24 clock switches to 'Optical.'
- To use an external device as the master (generating ADAT Sync, SMPTE timecode, word clock, or video sync), set Dakota's Clock Source to the corresponding input, and set all the Tango24 clock switches to 'Optical.'

Timecode sync

Timecode sync enables two or more devices to start at the same position and remain in sync as you record and play back. Timecode sync is useful if you want to record audio from one device to another for editing, and then transfer the edited audio back to the same position on the original device. It's also useful if you want to record on two separate devices simultaneously and then mix down from both.

Timecode information is transmitted in SMPTE, MTC (MIDI Time Code) and ADAT sync formats. See "Timecode Signals" later in this section for details about each format.

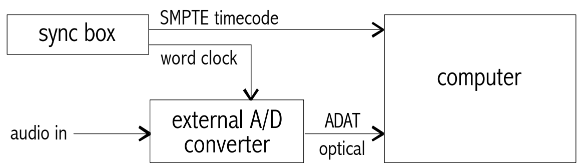
Timecode sync is not needed for hard-disk recording on only a single computer (for example, if you're recording multichannel audio through a Tango24 converter box connected to Dakota or Montana). Timecode sync is also not needed if you want to record digital audio from one device to another without transferring it back to the same position on the original device (for example, if you're recording from an ADAT tape machine to the computer, and then mixing to DAT).

As stated in the previous section, audio clock sources specify only a rate (samples per second). Timecode sources specify both a rate (in frames per second or fps) and a position (in hours, minutes, seconds, and frames). Here are the four standard frame rates, and their common uses:

- 24 fps – Film worldwide
- 25 fps – PAL/SECAM video in Europe
- 29.97 fps – NTSC color video in North America and Japan
- 30 fps – NTSC black/white video, and most music-only projects

Usually, the audio clock master and timecode source are the same device. When timecode is sent from the master device, the slave devices chase (or follow) the timecode so all devices are locked to the same position and operate at the same sample rate.

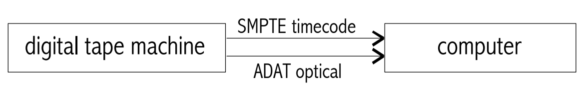
Sometimes, however, you may want to use an audio clock source that is different than the timecode source. In such situations, the clock and timecode sources must be referenced to a common timebase, as shown in the following example.



The sample rate (audio clock) in the ADAT optical signal and the frame rate (timecode) in the SMPTE signal must be locked together. Otherwise, the digital audio system doesn't know which rate to follow.

A classic problem occurs when "wild" SMPTE timecode is recorded on a digital tape machine. "Wild" means that the SMPTE generator was not locked to the digital tape machine's sample rate while recording the timecode and audio.

When the audio is digitally transferred to the computer using SMPTE timecode sync, the computer sees two different rates. Some audio applications try to adjust the audio to match the SMPTE timecode rate, but that taxes the computer's CPU and can introduce audible distortion.



Dakota sync features

In an integrated audio/sync system such as Dakota, the sample rate can be locked to a timecode input. If a 30fps SMPTE timecode signal is the source for a 44.1kHz sample rate, Dakota can lock the audio clock to the timecode source by ensuring that there are 44100 audio samples for every incoming 30 timecode frames — particularly useful when locking to a video deck with SMPTE output, or when recording tracks from analog tape that has SMPTE timecode recorded on it.

Systems without integrated audio/sync functions try to manipulate the audio data to obtain the correct sample rate. Some add or remove samples; others perform crude sample rate conversions. However, these techniques tend to distort the resulting audio, slowing down the computer. Dakota's hardware-based chase-lock capabilities eliminate those sources of audio distortion, and free the CPU for other tasks.

— Sample-accurate sync —

Dakota supports sample-accurate transfers between the computer and an ADAT machine, as shown in this example using Cool Edit Pro.

1. Connect the ADAT's 9-pin sync output to Dakota's ADAT 9-pin sync input, and set Dakota's control panel options to:

Clock source = ADAT sync Sample rate = 48 kHz Timecode source = ADAT sync Frame rate = 30 ND	Also, enable checkbox 1 after 'MIDI In devices (to software)' so Dakota port 1 will send ADAT-derived MTC to Cool Edit.

2. Open a new 48kHz project in Cool Edit Pro, and select multitrack mode.
3. In the Options menu, enable 'Sample Accurate Sync' and (under Settings→Devices) set MIDI IN to 'Dakota 1.'
4. Arm track 1 in Cool Edit Pro, and press F7 to enable SMPTE Sync mode.
5. Play the ADAT. Cool Edit Pro automatically starts recording (and displays waveforms) when it receives timecode.
6. Press F7 again to disable SMPTE Sync mode. Recording stops, and the waveforms are redrawn in their sample-accurate positions on the timeline.

— SMPTE timecode —

Using Dakota's unique SoDA™ technology (SMPTE on Digital Audio), Dakota and Montana can receive SMPTE timecode on any audio input, and transmit SMPTE timecode on any audio output. See 'Timecode Source' in the "Software Reference" chapter for detailed instructions.

The Dakota control panel lets you set the SMPTE output level and specify which ports are used for SMPTE timecode. See the "Software Reference" chapter earlier in this Guide for more details.

Setting the correct SMPTE level used to be difficult on old analog tape decks. If the level was too low, the signal would be lost in the noise. If the level was too high, the signal would bleed onto neighboring audio tracks. Cross-track bleeding is greatly reduced in modern digital systems, so Dakota's default SMPTE level works quite well.

Timecode Signals

SMPTE – Audio engineers commonly use the term "SMPTE" when referring to Linear Time Code (LTC) as defined by the Society of Motion Picture and Television Engineers (SMPTE). A SMPTE signal is a series of alternating tones which can be transmitted and recorded just like any other audio signal. It encodes time as a series of frames, each containing 80 bits of data including frame boundaries, the time (expressed in hours, minutes, seconds, frames), as well as several user-definable bits. The timecode portion is displayed as a four-part number such as 10:37:09:12, meaning 10 hours, 37 minutes, 9 seconds, and 12 frames.

MTC – The Musical Instrument Digital Interface (MIDI) specification includes a method of transmitting and receiving timecode using MIDI messages. MIDI Time Code (MTC) is also based on frames. One MTC message is transmitted for every quarter frame of timecode, so the resolution of MTC at 30fps is fixed at 8.33 msec (1/30 x 1/4 = 0.00833).

ADAT Sync – Some ADAT-compatible devices, including Dakota and Montana, have 9-pin connectors for transmitting and receiving ADAT Sync signals. An ADAT Sync signal includes both rate (word clock) and timecode information, but the timecode is not frame-based. Instead, it identifies the timecode position to the resolution of the audio samples themselves, providing the capability of sample-accurate transfers — about 400 times the accuracy of SMPTE timecode or MTC.

Tuning the System

Because Windows 95 and Windows 98 are general purpose operating systems, some of the default system settings may not be optimal for digital audio work. This chapter offers suggestions that may help improve the performance of your system.

Please check your audio application's documentation for additional performance tips. If you have Internet access, you may want to periodically check your software manufacturer's web site in addition to ours (http://www.FrontierDesign.com) for updated information.

Disk Drives

Whenever possible, use a large, fast, separate disk for storing your digital audio. Digital audio consumes a lot of disk space. For example, 5 minutes of an 8-channel, 44.1kHz, 16-bit recording consumes about 212 MB on your hard disk. With disk prices decreasing all the time, large disks are quite affordable. Nevertheless, you need to be sure your disk has enough room for your projects. You may need to archive older projects to tape or removable disks.

The number of tracks that your software will be able to transfer in real time depends on the speed of your hard disk subsystem (and your CPU, of course). The EIDE disk subsystem that comes with many desktop systems should be fine in most cases but for the best performance, consider a PCI-based Wide UltraSCSI adapter and "AV" (audio-visual) rated disk drives with at least 7200rpm spindle rates.

Use a separate disk drive for your audio. By keeping your programs, virtual memory swapfile, and any disk caches on a different drive, you keep the disk activity associated with those operations separate from the continuous reading and writing of audio data. This should result in less head movement and higher overall throughput for your audio.

Thermal Recalibration

Don't use disk drives that interrupt data flow for "thermal recalibration," a compensation for mechanical changes that occur as the drive warms up and cools down.

We believe that most new drive designs (even non-AV rated drives) don't need thermal recalibration, but if you have an older drive and are experiencing audio dropouts under heavy loads (lots of tracks), you may want to ask the drive manufacturer whether thermal recalibration is performed on that type of drive.

Data Compression

Don't use data compression on the drive that stores your digital audio. Although Windows 95/98 supports data compression (for example, the DriveSpace utility that comes with Microsoft's "Plus!" add-ons for Windows 95), the compression algorithms generally don't work well with digital audio. Furthermore, because the CPU spends time compressing and uncompressing your files, fewer CPU cycles are available for mixing tracks in your audio application.

Disk Fragmentation

We recommend that you defragment your disk regularly, especially after heavy editing sessions. Defragmenting reorganizes files into contiguous areas on the disk, so that the entire file can be accessed with a minimum number of head movements. One strategy is to leave the PC running and defragment the disk at the end of each day.

Sporadic Disk Activity

Avoid applications and utilities that cause sporadic disk activity. Some applications think they're doing you a big favor by hanging around in the background until the last crucial 10 seconds of your mix-down, when they awaken and start pounding your disk.

Here's a list of some applications and utilities that cause sporadic disk activity, and suggestions for thwarting them.

— System Agent —

System Agent (included in the Microsoft Windows add-on) includes utilities that scan your disk for errors and perform disk defragmentation. System Agent allows you to schedule these activities so they occur at specific times or when you haven't used your computer for a period of time. To prevent these disk-intensive activities from occurring, we recommend that you either disable System Agent while doing your audio work (right-click its icon in the system tray of the Task Bar), or schedule the activities to occur when you're absolutely sure that you won't be trying to record and play audio.

— Find Fast for Office —

If you use Microsoft's Office products (Word, Excel, PowerPoint, Schedule, or Access), the "Find Fast" utility was probably installed in the Windows Control Panel. The utility periodically scans your disks, building indices of all your Office documents so their names can be displayed quickly. Unfortunately, the indexing activity could pound your disks while you're trying to play back a multitrack recording.

To prevent Find Fast from running, you can either remove it permanently or disable it temporarily while you work on audio.

- To disable Find Fast permanently, open the Windows "Startup" folder and delete "Microsoft Office Find Fast Indexer." (You can use Windows Explorer to navigate to "C:\Windows\Start Menu\Programs\StartUp.")
- To disable Find Fast temporarily, open the Windows Control Panel (Start→Settings→Control Panel) and double-click the Find Fast icon. When the Find Fast window appears, select 'Pause Indexing' from the Index menu. When you want to enable Find Fast, select 'Pause Indexing' again.

— Screen Savers —

Few things are worse than having your audio interrupted by the sudden appearance of a bunch of flying toasters (or some other screen saver). A screen saver is particularly vexing because its delay timer silently ticks away when you're not touching your mouse or keyboard (when you're printing your final mix to DAT, for example).

To disable the screen saver, consult its documentation, or try this procedure: open the Windows Control Panel, double-click the Display icon, select the "Screen Saver" tab, and click the down arrow next to the "Screen Saver" box. Select 'None' (the first entry) and then click 'OK' to confirm the selection.

— CD-ROM "Auto-Insert" Notification —

When this option is enabled, Windows may periodically poll your CD-ROM drive to see if you've changed CD's, causing the CD-ROM access light to blink about every five seconds. If Windows Explorer is open and detects a new CD, you'll experience a flurry of disk and CPU activity while the folder and filenames of the new CD are updated in the display. The CPU activity may cause dropouts in your audio.

If you don't load a CD during audio operations, you may not experience this problem, but it's a nice opportunity to rid your machine of that annoying flashing LED!

1. Open the Windows Control Panel (Start→Settings→Control Panel), and double-click the System icon.
2. When the "System Properties" dialog box appears, click the "Device Manager" tab and then click '+' to open the "CD-ROM" sub-tree. A list of installed CD-ROM drives appears (there may be only one in the list).
3. Double-click the line that describes your particular model of CD-ROM ("NEC CDR-74" for example).
4. When the "Properties" dialog box appears, click the "Settings" tab and uncheck 'Auto insert notification.'
5. Click 'OK' to confirm the change, and then close the Control Panel.

Multi-tasking

Windows 95/98 can run many applications at once, but if you run more than one at a time, the performance of each individual application diminishes. Your audio applications are particularly heavy users of CPU cycles and disk bandwidth, so it's important to minimize the use of other high-demand applications. It's probably OK to run a word processor in the background while you're mixing down, but avoid applications that do a lot of computation or frequently access the hard disk.

File Caching

Windows 95/98 has built-in file caching which saves your most recently accessed data in memory. Since file caching is of limited use in digital audio applications, you may want to restrict the amount of memory reserved for file caching. Locate the SYSTEM.INI file in your Windows folder, and open the file in a text editor such as Notepad. Create (or find and edit) a section named "[VCache]" and make sure the MinFileCache and MaxFileCache are both set to:

4096 (for a computer with 32MB of RAM), -or-
8192 (for a computer with 64MB or more of RAM)

The values are given in kilobytes (KB) of RAM (2048 means 2MB of RAM).

> WARNING! Do not use values for MinFileCache or MaxFileCache that begin to approach the amount of physical RAM in your machine. NEVER use values greater than the old Windows 3.1 limit of 24576 (24 MB). Microsoft specifically warns that it does not guarantee performance of the cache at sizes larger than this and disavows responsibility for data corruption or loss that may occur. So do we!

Other Caching Options

There are several other caching options which may affect your PC's audio performance —

- You may want to limit the amount of "Read-ahead optimization" performed by Windows. Open the Control Panel (Start→Settings→Control Panel), and double-click the System icon. Select the "Performance" tab, and click the 'File System...' button. When the "File System Properties" dialog appears, select the "Hard Disk" tab, and move the 'Read-ahead optimization' slider all the way to the left.
- In the same dialog ("File System Properties"), you can specify the "Typical role of this machine." In most cases, you'll want it set to 'Desktop computer.'
- You may want to disable "write-behind caching." In the "File System Properties" dialog, select the "Troubleshooting" tab and check the box labelled 'Disable write-behind caching for all drives.'

Virtual Memory Swapfile

When the demands of multi-tasking exceed the limits of physical memory on your machine, Windows dynamically changes the size of the swapfile that it uses to hold code and data. You can improve audio performance by setting the swapfile parameters manually.

Open the Control Panel, double-click the System icon, select the "Performance" tab in the System Properties dialog, and click the "Virtual Memory…" button. When the "Virtual Memory" dialog appears, select "Let me specify my own virtual memory settings."

Set the following parameters in the dialog:

Hard disk: choose a disk other than the one you use for digital audio (if possible)
Minimum: set it to 2.0–2.5 times the size of your system memory
Maximum: set it to the same value you specified for minimum

Leave "Disable virtual memory" unchecked!
(Otherwise, Windows displays a warning.)

Click "OK" to confirm your choices, and close the Control Panel. After changing this setting, you'll be reminded that the new settings won't take effect until you restart Windows.

Graphics Acceleration

If you experience glitches during recording or playback, you may have a video driver problem. If so, try reducing the amount of graphics acceleration used by Windows. Open the Control Panel, double-click the System icon, select the "Performance" tab in the System Properties dialog, and click the "Graphics…" button. When the "Advanced Graphics Settings" dialog appears, move the "Hardware acceleration" slider all the way to the left. Click "OK" to confirm the setting, and close the Control Panel. After changing this setting, you'll be reminded that the new settings won't take effect until you restart Windows.

After you've rebooted, see if the glitching problem is gone. If so, you may want to experiment with other settings of the "Hardware acceleration" slider to regain some of the graphics performance.

32-bit Drivers

To obtain the best possible performance from your disk system, use 32-bit drivers. Generally, the Windows installation replaces real-mode drivers with 32-bit protected mode drivers. However, there may not have been any 32-bit drivers that were compatible with your hardware when Windows was released. To check the driver, open the Control Panel, double-click the System icon and select the "Performance" tab. If the "File System" status doesn't indicate "32-bit," contact the manufacturer of your disk controller to get a 32-bit protected mode driver and install it in your system.

Windows System Monitor

Windows includes a "System Monitor" utility that displays important performance information about your machine. If it's installed, you can usually find it by following this path in the Start menu:

Start→Programs→Accessories→System Tools→System Monitor

What if you don't have System Monitor?

System Monitor is optionally installed as part of Windows 95/98, but every Windows 95/98 PC can load it. Open the "Add/Remove Programs" control panel (Start→Settings→Control Panels→Add/Remove Programs). In the "Windows Setup" tab, scroll down and select "Accessories." Its check box will be gray if some of the accessories aren't installed. Click the 'Details…' button. Scroll through the list of programs and make sure "System Monitor" is checked. Click 'OK' and follow the installation instructions.

When Dakota's driver is loaded, it notifies Windows that it can collect statistics that indicate how well the system is responding to Dakota's requests to read and write audio data.

You can display these statistics and compare them with other important performance data —

1. Start the System Monitor application (Start→Programs→Accessories→System Tools→System Monitor).
2. Click on the fourth icon in the tool bar (it looks like a funky blue mountain range) so that data will be displayed in "time-line" format.
3. Select "Add Item…" from the Edit menu.
4. In the "Category" box, click 'Dakota Performance.' The list on the right indicates which Dakota performance data can be displayed.
5. Select one or more items from the list and click 'OK.' The graphical display adds the statistics you selected. You may also want to select items from other parts of the system so you can correlate Dakota's performance with other CPU and disk activity.

The following list describes the available Dakota statistics.

— IRQ Latency —

This statistic indicates the maximum number of audio samples that have elapsed between the time the Dakota hardware asked for service and the time Windows began to process the request. Typically it's less than 5.

Correlating the system's response time to interrupts with other system activity will help you determine if there are other installed devices that prevent Dakota from being serviced efficiently. To calculate the absolute latency (accurate to within one sample), divide the IRQ latency by the current sample rate.

— IRQ Misses —

Normally, this should be zero. An IRQ miss means the Dakota driver wasn't given enough time to transfer audio between your PC and the Dakota hardware. This is a severe problem, and indicates that something in the system is causing gross timing latencies that can't be overcome. The result can be brief audio dropouts or pops. Try to determine whether another device is active when the IRQ misses occur, and then correct the situation by removing that device or by not using it while also trying to transfer audio.

— Input Overruns —

This statistic indicates how many errors occurred when Dakota was ready to transfer newly-recorded information to an application, but the application hadn't supplied a buffer in which to write the audio data. Input overruns are generally caused by an overloaded CPU or a disk system that can't keep up with the audio applications.

— Output Underruns —

This statistic indicates the number of times that the driver ran out of data to send to the digital audio outputs. Generally, it means that the CPU or disk system couldn't keep up with the real-time demands of the application. It's normal to see a "blip" when playback stops, but an output underrun in the middle of playback indicates a problem.

≡ Dakota Specifications ≡

These specifications are subject to revision without notice.

Digital audio inputs	16 channels ADAT optical format (2 TOSLINK optical ports) 2 channels SPDIF format from CD-ROM digital output, or either TOSLINK optical port, or coaxial input (RCA phono connector via breakout cable from HD-15 port)
Digital audio outputs	16 channels ADAT optical format (2 TOSLINK optical ports) 2 channels SPDIF format on both TOSLINK optical ports and on RCA phono connector via breakout cable from HD-15 port
MIDI I/O	2 MIDI inputs + 2 MIDI outputs via breakout cable connected to miniDIN-8 port (can be expanded to 8 MIDI inputs + 8 MIDI outputs + SMPTE I/O with Sierra MIDI/SMPTE box connected to Dakota's miniDIN-8 port)
ADAT sync input	DB-9 connector via breakout cable from HD-15 port
SMPTE timecode	Any audio input or output can be used for SMPTE timecode; Dedicated SMPTE timecode I/O jacks available via Sierra
Sample rates	44.1 and 48 kHz internal, Varispeed and input tracking from 39–51 kHz, Real-time resampling to support 8, 11.025, 16, 22.05, and 32 kHz output streams
Audio data	8, 16, 20, 24 bit resolution
Software	Windows 95/98 device driver, Control panel for displaying status and setting options
System requirements	1 PCI slot and 1 IRQ
Internal expansion	40-pin connector for cable to Montana board
Dimensions	6.6" x 4.2" (16.8cm x 10.7cm), PCI short card

FC Compliance Statement

This equipment has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the User's Guide, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment to an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

The user is cautioned that changes and modifications made to the equipment without the approval of manufacturer could void the user's authority to operate this equipment.

CE Declaration of Conformity

Frontier Design Group, LLC declares that the multichannel digital I/O card called Dakota conforms to the following Directives and Standards:

Council Directives: 89/336/EEC, 73/23/EEC
Conformance Standards: EN55022 Class B, EN50082-1

Manufacturer: Frontier Design Group, LLC
199 Heater Road
Lebanon, NH 03766 (USA)
603-448-6283 voice
603-448-6398 FAX

European Contact: Et Cetera Distribution
Valley House
2 Bradwood Court, St. Crispin Way
Haslingden
LANCS, BB4 4PW, UK
+44 1706 228039 voice
+44 1706 222989 FAX